Amendments to the Specification:

Please replace a paragraph beginning at line 4 on page 3 of the above-identified application with the following amended paragraph:

One example of a software tool for use in a personal computer is an audio conferencing tool described in "vat--LBNL Audio Conferencing Tool", published May 1996 and available at ~~http://wwwnrg.ee.lbl.gov/vat~~ wwwnrg.ee.lbl.gov/vat. The packets generated by this tool conform to the Real-Time Transport Protocol (RTP) as described in "RTP: A Transport Protocol for Real-Time Applications", Network Working Group, January 1996, which is available from the Internet Engineering Task Force (IETF) website as Request for Comment (RFC) 1889, ~~http://www.ietf.org/rfc/rfc1889.txt?number=1889~~ www.ietf.org/rfc/rfc1889.txt?number=1889. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services.

Please replace a paragraph beginning at line 25 on page 8 of the above-identified application with the following amended paragraph:

One goal of the system administrator of a computer network is to limit unauthorized outside access to the computer system. The system administrator starts with a completely closed system (i.e., no outside traffic is allowed in) and opens up a number of ports to allow access for web traffic and Telnet access. TCP port 80 is the port number commonly assigned by system administrators for incoming web traffic. Other port numbers may be used for File Transfer Protocol (FTP) and Telnet access. The system administrator obtains these commonly assigned port numbers from industry guidelines which suggest which port numbers should be used for a particular use. The Internet Assigned Numbers Authority (IANA) houses the many unique parameters and protocol values necessary for operation of the Internet and its future development. These parameters and protocol values may be found at ~~http://www.iana.org/numbers.htm~~ www.iana.org/numbers.htm. Types of numbers

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range from unique port assignments to the registration of character sets. A copy of the list of port numbers may be found at ~~http://www.isi.edu/in-notes/iana/assignments/port-numbers~~ www.isi.edu/in-notes/iana/assignments/port-numbers. It is from these lists that port numbers were obtained for use in conjunction with an embodiment of the invention, as seen below.

Please replace a paragraph beginning at line 19 on page 11 of the above-identified application with the following amended paragraph:

In accordance with one embodiment, the call control mechanism is based on Request for Comments (RFC) 2543: "Session Initiation Protocol" (SIP) (March 1999) developed by the Internet Engineering Task Force (IETF), an international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture. SIP discloses an application-layer control (signaling) protocol for creating, modifying and terminating sessions with one or more participants. These sessions include Internet multimedia conferences, Internet telephone calls and multimedia distribution. Members in a session can communicate via multicast or via a mesh of unicast relations, or a combination of these. SIP invitations used to create sessions and carry session descriptions which allow participants to agree on a set of compatible media types. SIP supports user mobility by proxying and redirecting requests to the user's current location. Users can register their current location. SIP is not tied to any particular conference control protocol. SIP is designed to be independent of the lower-layer transport protocol and can be extended with additional capabilities. RFC 2543 is available at ~~http://www.ietf.cnri.reston.va.us/rfc/rfc2543.txt?number=2543~~ www.ietf.cnri.reston.va.us/rfc/rfc2543.txt?number=2543 and discloses a conventional method of transmitting destination port information over call control. This method may also be modified to transfer source port information of a sender, as described herein.

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Please replace a paragraph beginning at line 19 on page 11 of the above-identified application with the following amended paragraph:

Port 5004 is recommended by the RTP RFC standard for UDP transmission, RFF 1890, RTP Profile for Audio and Video Conferences with Minimal Control, January 1996, ~~http://www.ietf.org/rfc/rfc1890.txt~~ www.ietf.org/rfc/rfc1890.txt. However, as stated above, the chosen number does not matter as long as the number is fixed across the private branch exchange system. The User is usually not authorized to change this number. The firewall administrator is usually the only one authorized to open a hole at this chosen port number.

Please replace a paragraph in the Abstract section, beginning at line 5 on page 53 of the above-identified application with the following amended paragraph:

~~A new method to limit the number of holes a system administrator must open in a firewall for internet telephony.~~ The number of holes that are opened in ~~the~~ a firewall for internet telephony is limited to a first hole used for call control and a second hole used for audio traffic. Fixed destination ports for telephony traffic and call control traffics are created at a destination. Media streams are received at the telephony fixed destination port. The source of each media stream is commanded to provide a unique identifier for each media stream arriving at the destination from each source. Each media stream is identified by a unique identifier provided by the source. The unique identifier for each media stream is communicated to the destination by each source over call control. All telephony traffic is received only at the fixed destination port for telephony and all call control is received only at the fixed destination port for call control. ~~The use of fixed destination ports for all audio traffic and all call control traffic allows the number of firewalls in a system to be kept to a minimum.~~

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